REMARKS

By this Amendment, claims 11, 13, 15-16, 23 and 26 have been amended. New claims 30-59 have been added. Thus, claims 1-59 are pending in the present application.

As mentioned above, claim 23 has been amended to depend from claim 1, as was the case in the application as originally filed. In view of such, the invention recited in claims 23-26 of Group II each incorporate the features recited in claim 1, and therefore the basis for the restriction requirement is now moot.

Claim 16 has been amended to correct a typographical error, and new claims 30-59 have been added to provide specific protection for the preferred aspects of the invention originally recited in the claims at the time of filing and which were removed in the Amendment filed on September 6, 2001 and also in amended claims 11, 13, 15 and 26 herein.

Applicant respectfully submits that all of the pending claims 1-59 are patentably distinguishable over the prior art of record for the reasons discussed in the Remarks section of the Amendment filed on September 6, 2001, and further that the present application is currently in condition for allowance, whereupon early and favorable reconsideration in this regard is courteously solicited.

Respectfully submitted,

Ellen S. Tao

Ellen S. Tao
Registration No.: 43,383
OSTROLENK, FABER, GERB & SOFFEN, LLP
1725 K Street NW, Suite 1108
Washington, D.C. 20006
Telephone: (202) 457-7785

APPENDIX A

"CLEAN" VERSION OF EACH PARAGRAPH/SECTION/CLAIM

37 C.F.R. § 1.121(b)(ii) AND (c)(i)

CLAIMS (with indication of amended or new):

B1 SUB C4

11. (Amended) The method of claims 1, further comprising generating an energy signal which is proportional to an energy content of the ambient noise from said audio signal or from a signal derived from said audio signal.

B2 SUB C6

13. (Amended) The method of claim 12, wherein said second low pass filtering is effected digitally in the form of a convolution over 20 to 70 values.

B3

15. (Amended) The method of claim 11, further comprising performing a subsequent differentiation of the energy signal with respect to time to obtain an energy difference signal.

SUB C7

16. (Amended) The method of claim 1, wherein the range of normalized values D is defined by a lower limit D_u, and an upper limit D_o, and wherein the normalization is effected by:

- obtaining the maximum of the absolute value of the audio signal or the derived signal within the duration of normalizing the audio or derived signal, which is shorter than or equal to the duration of a hearing sample,
- multiplying the reciprocal value of said maximum by $(D_0 - D_u+1)$, and
- multiplying this product by each value of the audio or derived signal within the duration of the normalized signal.

B4

23. (Amended) Method for evaluating hearing samples processed according to claim 1, comprising:

recording a plurality of samples of programs to be monitored wherein the samples have at least the same duration as a corresponding plurality of hearing samples,

subjecting the program samples and the hearing samples respectively to the same processing steps, and

calculating a first correlation for comparing the hearing samples with the processed program samples in order to find a match.

B5

26. (Amended) The method of claim 24, wherein the comparison of the hearing samples with the program samples is effected in two passes, wherein a first pass comprises comparing a respective hearing sample to all program samples using said first correlation the calculation of which uses a coarse graduation of the time shift, and wherein a second pass comprises using a second, more rugged correlation which provides a finer graduation of the time shift which has a correlation value above a predetermined unit.

B6

30. (New) The method of claim 1, wherein the electroacoustic transducer is a microphone.

31. (New) The method of claim 3, wherein said mapped result is represented by binary numbers having a fixed number of binary digits from 4 to 8 bits .

Sub C1

32. (New) The method of claim 3, wherein said mapped result is represented by binary numbers having 4 bits of binary digits.

33. (New) The method of claim 4, wherein any content of the other band signals contained in said each band signal is attenuated to half of their respective original levels.

34. (New) The method of claim 4, wherein any content of the other band signals is completely attenuated from said each band signal so as to not be present at all therein.

35. (New) The method of claim 5, wherein said audio signal is divided into from 4 to 10 band signals.

SUB C4

36. (New) The method of claim 5, wherein said audio signal is divided into from 5 to 8 band signals.

SUB F1

37. (New) The method of claim 5, wherein said audio signal is divided into 6 band signals.

38. (New) The method of claim 7, wherein all first low pass filterings have a same Q-factor.

B4 cont

39. (New) The method of claim 8, wherein said low pass filtering is realized by means of a digital convolution over15 to 25 values.

SUB C12

40. (New) The method of claim 8, wherein said low pass filtering is realized by means of a digital convolution over 19 values.

41. (New) The method of claim 10, wherein n is equal to 2.

42. (New) The method of claim 11, wherein said energy signal is generated by squaring said audio signal or said signal derived therefrom.

SUB C13

43. (New) The method of claim 13, wherein said second low pass filtering is effected digitally in the form of a convolution over 40-55 values.

44. (New) The method of claim 13, wherein said second low pass filtering is effected digitally in the form of a convolution over approximately 48 values.

45. (New) The method of claim 13, wherein the convolution has coefficients which are essentially equal to each other.

46. (New) The method of claim 13, wherein the coefficients of the convolution are equal to 1.0.

S4B
C1
B6
cont

47. (New) The method of claim 14, wherein n is equal to the number of values of the convolutions of the second low pass filtering.

48. (New) The method of claim 15, wherein said differentiation is performed by computing the difference between two respective values of the energy signal.

49. (New) The method of claim 16, wherein D_o is equal to 0.

50. (New) The method of claim 16, wherein D_u - D_o is preferably equal to 2^n-1, n being a whole number greater than 4.

51. (New) The method of claim 50, wherein n is equal to 7.

52. (New) The method of claim 16, wherein the duration of normalizing the audio or derived signal is equal to the duration of a hearing sample.

53. (New) The method of claim 17, wherein essentially all steps of the method are performed by binary arithmetic with a number of digits as provided by a computing unit performing the method.

54. (New) The method of claim 20, wherein the timer switches off the hearing sample unit in the periods between the processing of two hearing samples, in order to reduce energy consumption.

55. (New) The method of claim 22, wherein the device is incorporated in the form of a wristwatch.

56. (New) The method of claim 26, wherein the second correlation is used in the case in which the time shift has correlation values c_t above a predetermined limit.

57. (New) The method of claim 26, wherein the second correlation provides a resolution of the time shift which is at least twice as high as that obtained with the first correlation.

58. (New) The method of claim 26, wherein said second correlation is chosen such that great deviations between the hearing and the program sample have a smaller influence upon the correlation coefficients than in the first correlation.

59. (New) The method of claim 26, wherein said second correlation is effected according to the formula

$$r_t = \frac{\sum_{i=1}^{N} |s_i - a * m_{i-t}|}{\sum_{i=1}^{N} |s_i|}$$

wherein

N	:	number of hearing sample values used in the correlation,
t	:	time shift between the hearing and the program sample,
S_i	:	hearing sample value at the time i,
m_1	:	program sample value at the time i, and
a	:	scaling factor which takes account of the damping of the program signal with respect to the hearing sample;
r_t	:	correlation value for the shift t, 0 (optimal correlation) < r_t < 1 (no correlation),

a being determined in such a manner that r_t assumes a minimal value.

APPENDIX B

VERSION WITH MARKINGS TO SHOW CHANGES MADE
37 C.F.R. § 1.121(b)(iii) AND (c)(ii)

Claims:

11. (Amended) The method of claims 1, further comprising generating an energy signal which is proportional to an energy content of the ambient noise from said audio signal or from a signal derived from said audio signal[, said energy signal being generated by squaring said audio signal or said signal derived therefrom].

13. (Amended) The method of claim 12, wherein said second low pass filtering is effected digitally in the form of a convolution over 20 to 70 values[, the coefficients of the convolution being essentially equal to each other].

15. (Amended) The method of claim 11, further comprising performing a subsequent differentiation of the energy signal with respect to time to obtain an energy difference signal[, said differentiation being performed by computing the difference between two respective values of the energy signal].

16. (Amended) The method of claim 1, wherein the range of normalized values D[,] is defined by a lower limit [D_u D_o] $\underline{D_u}$, and an upper limit $\underline{D_o}$, and wherein the normalization is effected by:

- obtaining the maximum of the absolute value of the audio signal or the derived signal within the duration of normalizing the audio or derived signal, which is shorter than or equal to the duration of a hearing sample,
- multiplying the reciprocal value of said maximum by $(D_0 - D_u+1)$, and
- multiplying this product by each value of the audio or derived signal within the duration of the normalized signal.

23. (Amended) Method for evaluating [recorded] hearing samples processed according to claim 1, comprising:

recording a plurality of samples of programs to be monitored wherein the samples have at least the same duration as a corresponding plurality of hearing samples,

subjecting the program samples and the hearing samples respectively to the same processing steps, and

calculating a first correlation for comparing the hearing samples with the processed program samples in order to find a match.

26. (Amended) The method of claim 24, wherein the comparison of the hearing samples with the program samples is effected in two passes, wherein a first pass comprises comparing a respective hearing sample to all program samples using said first correlation the calculation of which uses a coarse graduation of the time shift, and wherein a second pass comprises using a second, more rugged correlation which provides a finer graduation of the time shift which has a correlation value above a predetermined unit[, said second correlation being chosen such that great deviations between the hearing and the program sample have a smaller influence upon the correlation coefficients than in the first correlation, and being effected according to the formula

$$r_t = \frac{\sum_{i=1}^{N} |s_i - a * m_{i-t}|}{\sum_{i=1}^{N} |s_i|}$$

wherein

N	:	number of hearing sample values used in the correlation,
t	:	time shift between the hearing and the program sample,
S_i	:	hearing sample value at the time i,

m_i : program sample value at the time i, and

a : scaling factor which takes account of the damping of the program signal with respect to the hearing sample;

r_t : correlation value for the shift t, 0 (optimal correlation) < r_t < 1 (no correlation),

a being determined in such a manner that r_t assumes a minimal value].